A QUICK GUIDE TO AUDIO PRODUCTION

A practical guide to audio production

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Intro

Audio is a very big topic, and contributes to so many different industries. What I want to do here is give you audio the way I experienced it, in condensed form, so you, too, can grasp, in a short time, all the nuances that go into sound and audio technology. Along the way, I will be dispelling various things about audio, external to just the media production world.

There is so much to say on the topic of audio. When I first journeyed into audio, I did it for personal reasons. I used to go to Salzer's music store to buy tapes in the 90's when I was a kid. This led me to using tapedecks, and not too long after, full on multitrack machines and digital audio workstations: my first machine was a Zoom MRS 1266 multitrack recorder I got when I was fifteen. I will be explaining what all these are in just a minute, but as you can see, one thing led to another, and that is often what happens in audio. One thing leads to another thing, 2 then you have wires everywhere with an interesting idea in mind. I used to sit there for hours trying to figure out how a function worked, how to dim faders, how to use audio drivers, how to implement MIDI, or loop drum beats. It took years before I figured everything out fully, but with this manual in hand, you can read it, and actually grasp these things easily. I used to play around a lot and learn things on my own. The manuals that came with the equipment were very function oriented, and much less about theory. I want to give the basic theories, and the basic ideas behind all the functions you will come across. I hope you have a great time, and happy learning!

-Jeffrey Gates Toves

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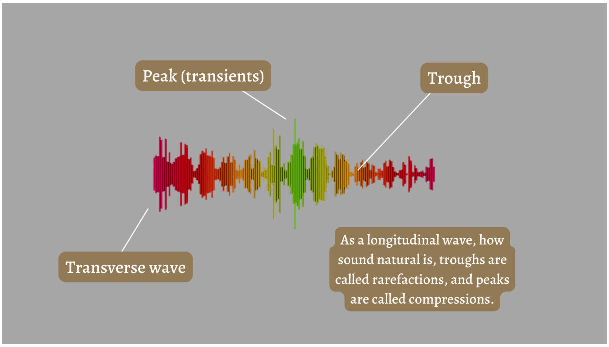
Sound

Chapter I

Sound

Sound

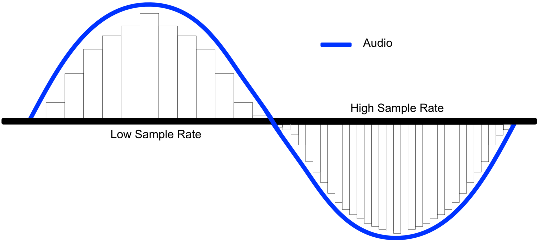
Sound is simply the atmospheric pressure created when sounds are made. It is air pressure that gets perceived by our ears. The human hearing range sits between 20 hertz and 20 kilohertz. Hertz is the unit of frequency, and this is cycles per second. 20 hertz is 20 cycles per second; 20,000 hertz, or 20 kHz (20 kilohertz), is 20,000 cycles per second. We hear sound through our ears, which go through the outer ear, into our ear drum, and like how an amplifier works, a hammer and anvil action mechanically vibrates our inner ear, which then gets perceived as sound outputted via the cochlea; the part of the ear that acts as a transducer and takes the vibrations of sound and translates it into electrical impulses.


Peak (transients)
Trough
Transverse wave
As a longitudinal wave, how sound natural is, troughs are called rarefactions, and peaks are called compressions.

Sound wave.

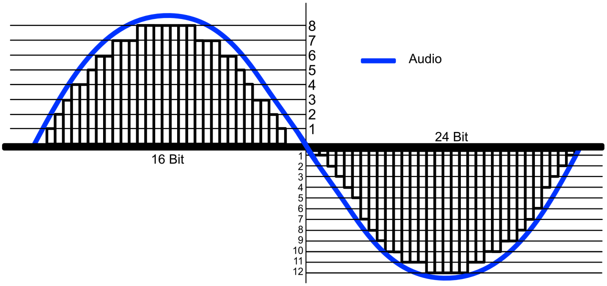
We hear sound in all its forms, through music, to a dog barking, etc. Some sounds are loud, and some not so loud. In the audio world, much of the sound that is unwanted is just called noise. Note: there are various types of these considering the application. Sometimes this is added to synthesizer sounds or effects to add certain qualities. There is a lot to know, so from here, we will move on.

Digital Audio



Samplerate graph.

Digital audio is exactly that, audio in digital format. It requires a computer to create, encode, and decode the waveform. Every waveform consists of two components, a samplerate, and a bitdepth. Samplerate is the number of samples used in a waveform; per second that is. Thus, 44.1 kHz means that 44,000 samples are used in one second of audio. Bitdepth refers to the other component of the waveform that gives logic to the amplitude, or ‘volume level,’ of each sample in the recording. This is because recorded sounds come in an infinite variety with subtle increases and decreases in sound volume.



Bitdepth graph.

Audio Technology

An audio workstation is similar to a multitrack recorder, but within a computer. It is essentially a piece of software to record sounds. There are usually several main components to every workstation no matter what program you are using. There exists usually a recording section with tracks, a mixer with a master channel and channels for assigning to tracks, and a settings section to change the various settings in the software, such as the audio settings.

Recording is often as simple as hitting the record button, but usually it takes going into the settings to enable input for your mic interface or preamp and output to your speakers. Audio drivers must be chosen also to record high quality audio: this is your

ASIO driver. At this point, getting some basic input should be the outcome of all this, but you might run into some problems, but that’s where the fun begins.

Audio Interface

There are many to choose from, and various ways of receiving input from a microphone into a computer. The old school way often was just plugging in an old dynamic microphone with the traditional quarter inch audio jack plugged into a mini jack adapter, then this was plugged into the microphone input into the computer. If one had access to a mixer, or a microphone preamp, it was a similar process, but this time, the microphone was plugged into those devices, then those devices plugged into the computer via the mini jack. These days, it is almost universal that some sort of USB interface is used. These are referred to as audio interfaces.

An interface usually has a simple function of allowing either quarter inch jack input and output, XLR, input and/or output, with options for outputting to headphones for monitoring performance during recording. It’s very simple. Often the situation is, even for the previously mentioned (older) devices, phantom power is needed to power the more high end microphones. Condenser microphones require this phantom power, which is DC current that is provided for the higher quality electronic components within the microphone. It is an all in one device for high quality audio processing, and usually delivers the highest quality audio, usually through USB or Firewire, which are traditional computer processing formats that deliver fast data routing.

Equalizers

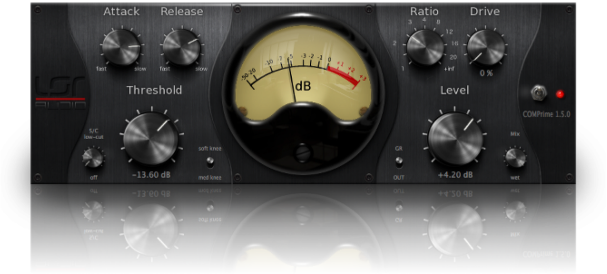
We can go to various areas of audio technology to get to understanding where and how equalizers are used, but simply speaking, equalizers provide a simple function in the audio world for corrections. Remember when we said the human hearing range is between 20 Hz and 20 kHz? Well, equalizers, whether they be digital types, or real hardware, they utilize this exact frequency range to allow the user to use the faders to change those particular frequency "bands." There are usually faders for low frequencies around 0-100 Hz, 100-300, etc, all the way to 20 kHz. Often there is also a frequency sweep option to allow for greater flexibility in choosing the frequency that one wishes to manipulate. This is referred to as "q," or quality factor, and enables the adjustment of bandwidth. A q above one is more selective in frequency, so narrower, while a q less than one is more wide. This helps with focusing in on a certain frequency so as to change the characteristics and level.

4 band outboard equalizer.

Stereos have them, audio tech has them. Actually, a lot of things have them. They all do this same thing. The hardware and software version usually is a simple piece of gear that enables audio to run through the equalizer via simple input and output. Graphics equalizers and parametric equalizers are the main types. They both serve the function of equalization, though with slightly different setups. Both equalizers can abstract away the bands into turnable knobs. This creates often a different style of equalization. Graphic equalizers are known to have a set of fixed bands with equal spacing, or ”octave spacing,” with a fixed bandwidth. Parametric equalizers allow for more precise control. In this type, you can control any band and change the bandwidth. We will learn more about this later, so let’s continue.

Compressors

In the world of audio, compressing things is one of the most important, even though repetitive, tasks audio guys and gals do in the field. Whether it be a drum kit, or just a kick, or snare, or an entire song, compression is used everywhere. It makes up a good amount of what music production is about. Simply what a compressor does is, it takes the peak values, or “transients,” of the audio and makes them smaller, or it makes the sound “compressed.”


Attack
Release
fast
slow
fast
slow
Threshold
-13.60 dB
S/C low-cut
off
soft knee
med knee
dB
Ratio
Drive
0 %
Level
+4.20 dB
GR
OUT
Mix
wet
COMPrime 1.5.0

Basic compressor in plugin format.

Every compressor, also known as a dynamic processor, has a few simple functions. These are: threshold, ratio, attack, release, and knee. The threshold is simply the level at which the compressor is processing the sound, and this works with the ratio. Every compressor technology out there will have this threshold and ratio section, thus this is often considered the soul of the compressor. What it does is it determines the amount of compression. For instance, a 2:1, or two to one, ratio means that for every 2 dB’s above the threshold that is set, it only allows 1 dB through. This essentially enables loud parts in music to be easier to listen and not be too startling or powerful so it is more streamlined with the rest of the production. A 4:1 ratio does the same sort of thing, as well as any ratio; for every 4 dB’s above the threshold, it lets in 1. This is very compressed, since if the threshold is at -20 dB, and peaks go above that by four dB’s, then only a small 1 dB gets through, which basically makes the sound quite “smashed.”

The attack and release are very straightforward functions on any compressor. The attack determines the amount of time it take for the compression to begin working, whereas the release is simply the time it take for the compression to stop working. The knee on the other hand is a function that is more related to how the signal is affected by a soft versus hard knee. This deals with how the compressor transitions between the compressed versus uncompressed sound, in considering all the “compressor” functions we went through.

The last topic here I want to mention is the idea of a limiter. Limiters are also dynamic processors, or a function within them, and dictates with the threshold just like other compressors do. However, for different reasons. These limit, or create a ceiling so sound cannot go beyond this limit. Limiters are often used during the mastering process to limit any sound beyond a certain, often very high, threshold. This is so as to keep audio from getting too loud. There are many ways to use limiters.

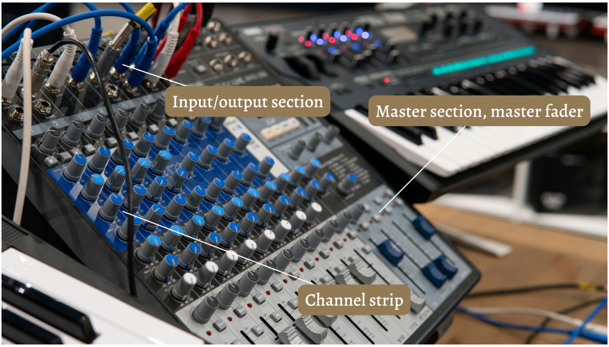
Delay

There are various effects in the world of audio, so I thought the best introduction to effects should be the most interesting of the bunch: delay. Delay can often be confused with echo, but it certainly is not echo. Delay is an audio production effect that can be emulated through either hardware or software. What it does is it takes in audio, through a delay unit, and 'repeats' the incoming sound with options to increase the rate of delay, or decrease it. Sometimes there are options for more fancy functions, such as spring type, reverb type, or multi delays. They are basically combined with other effects, or combined with multiple delay units that cross interact.

Delay in modern times can be synced to BPM in music sequencers, in both hardware and software, such as in Roland keyboards, Akai products, or in music software, such as Ableton Live or FL Studio. It is a useful function, else it becomes an applied situation where tempo matching is the case. Usually it is used this way in most basic uses, and it is just an increase or decrease in delay rate to manually tempo match the delay.

Mixers

Now to the next most fascinating topic in all of audio: mixers. And boy are they precious. That is because without mixers, there would be no way to really handle all the various signals coming in. It is the audio 'manager' that enables very advanced ways of dealing with audio. It is a singular piece of equipment that acts as a home for many modular components such as equalizers, compressors, amplifiers, etc.



Mixer functions.

Along the back of every hardware mixer is a line of amplifiers connected to each input. Thus, five inputs will equate to five channels, which each have five amplifiers, faders, equalizers. You get the point;

each channel has its own “effects strip.” Again, each input has its own personal amplifiers, eq, compressor, and fader, which culminates into a master fader that controls the volume levels of the combination of different settings at each channel. The master fader, its channel, has its own set of eq, compression, etc. Also notable functions on each channel are mute, on and off switches for the various channel effects, reverb, echo, etc.

Mixers are used for various purposes. Recording music is just one function if plugged into a multitrack recorder, or DAW. However, it is also used in churches, sound venues, radio stations, news and other television broadcast stations, movie production sets, public address systems, and the list goes on. It is very straightforward the technology, even its alternate uses. Often the case becomes about the included effects on each channel not being enough, thus requiring outboard effects or processors. Aux outputs are included on each channel enabling audio to go through an XLR or quarter inch, then through outboard gear, then back into another channel on the mixer. This is usually referred to as auxiliary routing. It is a big part of the audio industry, and it sure makes up a lot of how professionals create professional tracks.

DAW

In our attempt to uncover the great mysteries of audio, it is a must stop to get to grips with DAW's, also known as Digital Audio Workstations. These devices truly allow us to work with audio in a modern and professional way. Like previously mentioned, a DAW consists of several main components as an integrated piece of software. It usually has tracks, a mixer section, and a settings area for changing the software settings. Here we will want to dive into each part of a traditional DAW so we can use it well.

Setting up a DAW was briefly explained, but once audio has passed into the DAW, we can really get working. Audio in: this usually means that on each channel, there are options to choose which audio in, since you may have various inputs, such as from a guitar amp, a mixer, guitar mics, etc. Per channel, you should always make sure that you have chosen the right input, and this usually is from your audio interface.

Configuring the audio interface is an important part of working in a DAW. The interface provides high quality sound, and has its own audio drivers. The default driver in your computer is not fit to deal with the high quality sound coming from a professional audio interface, since latency and buffering issues can occur, which is a delay that exists in the digital audio world: just know low buffer rates equate to less latency, or lag. This lag is reduced in ASIO drivers, with the interface, or with a faster computer (processor). In your DAW settings in the audio section, you will have the option to change your driver to your interface or ASIO. Do this first, and you will never have problems with recording or using a DAW.

MIDI setup
Mic setup
DAW
Monitors

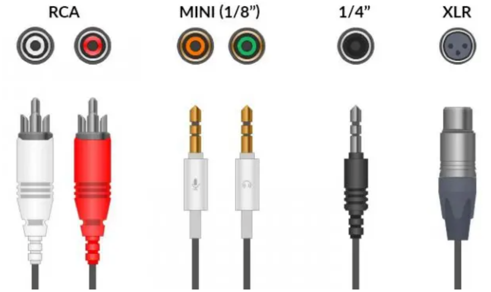
DAW setup.

The mixer part of a DAW is the same idea I mentioned previously. This is directly correlated to the tracks in the main window which lay on top of eachother for easy viewing and manipulation. Each track is allowed setting the track to its own channel on the mixer. When this is done, you can use plugins to enhance your recordings. Most DAW's have built in plugins, such as your basic effects: reverb, delay, echo, compression, phaser, chorus, flanger. Also included often are gain plugins, filters, distortion, more advanced dynamic processors and delays, mastering suites, and more.

We will be getting to recording in another chapter, but I hope this provided a good overview to get started. It's never too soon to experiment. A lot is gained when mistakes are made. Make them soon so you don't have to make too many. The reason for writing this book is so people can get up and running with the essential ideas so they can move on and create quality works. Taking the wonder out of this stuff can surely enable the grasping of more difficult areas in audio and music, and help with staying creative.

Cables

Cables can be a site, especially since this is where electronics becomes an actual part of the audio industry. We all got into it with an abstraction of the electronics, but we all surely scratched our heads, a lot, when dealing with cables. In and out seems simple until you're twenty connections in, not to mention any virtual connections, and it's all a tangled mess. Cables are either low noise or they're not. Low noise cables are necessary in recording and can be either XLR, which provides a stereo connection via three male prongs that connect to accepting female XLR inputs, or quarter inch, which has two main varieties. These are the TS and TRS varieties. The TS, or tip sleeve, has one ring near the tip, and these are used for "unbalanced" mono (one audio signal) signals, such as for electric guitars. TRS, tip ring sleeve, have two rings near the tip, and these are used for "balanced" signals, or stereo (two audio signals, left and right).



Common audio cables.

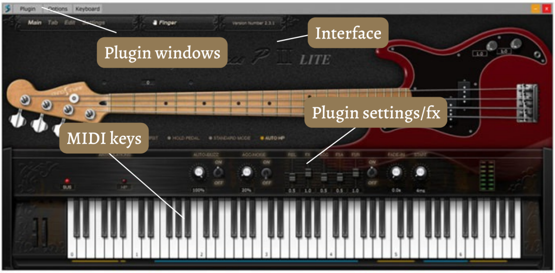
Balanced cables versus unbalanced cables have subtle differences, but have a big impact depending on the application. Unbalanced cables have two wires; a ground wire to prevent noise, and one audio signal wire, where sound from one devices goes through directly into another through this cable. Balanced cables on the other hand, or low noise cables, add an extra wire which duplicates the audio signal wire, creating two identical audio lines that are slightly out of phase. One of the audio wires is out of phase with the other, and is an inverted copy which gets put back into phase at the end. The ground wire, like unbalanced cables, help prevent noise, such as from electromagnetic interference.

XLR cables is the to go choice for most audio recording applications, since these are often balanced, stereo audio cables that transmit high quality sound with low noise. It uses three pins, with three pin input female connections on audio equipment, and three pin output male connections on cables. The three pins are for positive, negative, and ground, and is a very straightforward technology. These shouldn't be confused with DMX connection used for lighting. Though they have similar functionality, the internals are a bit different. Sometimes you will hear XLR as a catchall term for plugs of this type, especially for the mini XLR, which is often used to save space on small equipment.

Plugins

This area of audio is one of my favorites, since early on, most people get introduced to the fascinating array of plugins out there for effects, instruments, software, etc. It often is just too complicated at first, but I hope after this book, enough knowledge will be distilled so as to understand the complexities. Read the manuals, too. The manuals can give the subtle differences between similar technologies.

There are two common types of audio plugin, and these come in the VST/VSTi form (Virtual Studio Technology) or DX form (Direct X). Often plugins come in multiple formats, and is just about your preferred choice. There also are plugins for particular technologies such as AVID Protools's RTAS (Real-Time AudioSuite) and TDM (Time-division Multiplexing) plugins. These are designed to run with specially trademarked versions of AVID software.


Plugin
Options
Keyboard
Main
Finger
Interface
Plugin windows
LITE
Plugin settings/fx
MIDI keys

VSTi plugin instrument.

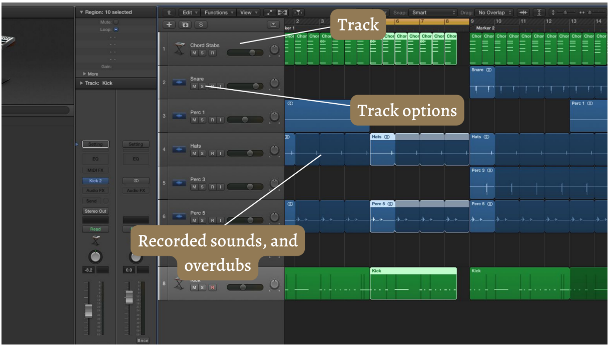
Plugins are used for additional effects and instruments in your DAW. There often is a limited amount of plugins in your DAW, thus we use plugins we buy from 3rd party sources. There are also many free plugins around the internet. Basically, these are small programs that act as modular inputs to your DAW, which enables direct integration into your studio environment. These are often pre tempo synced, and all it takes is adding the plugin onto a mixer channel, or as a track in your DAW or sequencer. Within the plugin can be anything, but they can include effects such as delay, reverb, echo, eq, and/or instruments, such as synths, pianos, and guitar sounds. Drum machines, too, are a very popular form of VST, as they often come in both standalone and plugin versions. Plugins are a great way to produce very high quality productions, and I recommend you learn a few, and find a routine for utilizing this technology in your workflow.

Track

I figured I'd include briefly about tracks and what they are. Software, and even hardware, come with a certain amount of default tracks. In software, these can be unlimited, but in hardware, there is often a set amount of tracks, such as five, as in a "five track recorder," but virtual tracks are usually an option with modern multitrack recording gear. Virtual tracks are just tracks beyond the amount of physical tracks on the recorder, and these virtual tracks operate similar to a DAW, and each virtual track just gets assigned hardware to use to record the virtual track.

In songs, voice recordings, such as in podcasts, or for film, tracks can grow into many tracks. This can be pretty complicated, especially with so many additions, cuts, and pasted tracks. I would like to give a few brief pointers on how to keep everything clean.

Each track recording is movable on the audio timeline. Where it sits is where it has any utility as a waveform, thus this makes these tracks susceptible to accidental adjustments. Each track has the option of locking the track on the timeline, and this so as things get more complicated, we can manage the recording session.


Region: 10 selected
Edit
Functions
View
Snap: Smart
Drag: No Overlap
Marker 2
Chord Stabs
Snare
Perc 1
Hats
Perc 3
Perc 5
Kick
Track: Kick
Setting
EQ
MIDI FX
Kick 2
Audio FX
Send
Stereo Out
Read
-8.2
0.0
Bnce

Track

Track options

Recorded sounds, and overdubs

Multitrack session.

Cutting and pasting is also a traditional editing component in the audio industry. A lot of time is spent on this. Sometimes a track needs to be cut somewhere and deleted, and then recorded from that point on. This takes stealth, and so this makes cutting more like a “scissors kind of thing.” It really is, because zooming in on the waveform enables very precise cuts within milliseconds of any recorded sound. It is very magical, and upon grasping this power, cutting and editing becomes very useful as a tool in your editing workflow.

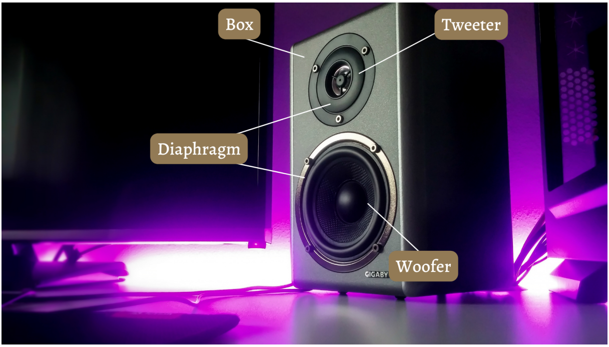
Crop is a special tool in editing and is used for selecting useful audio and deleting unneeded audio. This is useful in scenarios where there is a sizable amount of extra audio at the beginning and end of a recording, and what you want is just a clean version of the recorded "piece," or useful material. It is also helpful in many other scenarios, such as keeping parts that are wanted in general, such as from layered material and cutting the rest, or cutting loops in half, and many other things.

Amplifiers

Amplifiers are a complicated and lengthy subject, but it goes a long way talking about them for a bit. Amplifiers are traditionally electronic components. In the world of audio, they are used in guitar amplifiers, preamplifiers for mixers, audio interfaces, and recording consoles, "mic pre's," speakers (or "monitors"), and the list goes on. Essentially what an amplifier does is it takes an input source, of audio, and adds power, in watts, to the audio source and outputs it, which increases the volume. In audio interfaces, this is all done for you, as well as in all functional audio equipment.

Electronics sticks its complicated head out again in this area, and again we are faced with an electronics challenge. When an input source's energy is increased in watts, and becomes amplified, this signal will have a certain output impedance, or resistance, measured in ohms. A common understanding runs across the audio industry about this, and that's about impedance matching, or amplifier and speaker matching. If a speaker has an impedance of 8 ohms, and an "amp," or amplifier, has an impedance rating of 8 ohms, this is saying that the speakers match with the amplifier, since the speaker impedance is the same as the amplifier impedance. It usually is no more complicated than this for simple applications, but this is just a brief overview, and can get complicated fast. For instance, your amplifier should be rated in watts twice the amount than your speakers: 400 watt amp for 200 watt speakers. This all means that your amplifier should be able to deliver 400 watts per channel, or per speaker, into 8 ohms. If your amplifier is not matched perfectly, it's recommended your speaker impedance be a bit higher than your amplifier's impedance rating. Sensitivity in speakers, or "monitors," which we'll get to in a minute, is another useful piece of information related to all this. Sensitivity determines the ability of your speakers to get "loud." The higher the sensitivity, the better the loudness, measured in decibels per 1 watt per 1 meter, if you'd like to know. It is extremely important in the realm of matching. It is also good knowledge if you are looking for decent speakers that aren't too high in cost, since sensitivity can be read off on the products to determine desired loudness.

Monitors

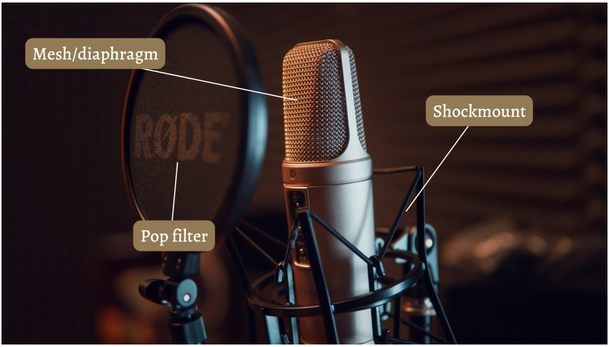


Parts of every monitor/speaker setup.

Here we are. We finally made it to speakers. This is surely getting interesting, since this is a very interesting and necessary part of audio as speakers are the all encompassing sources of sound throughout the world. I want to make this very brief, since I do not wish to complicate things here. Impedance matching is already understood, thus this leaves room for what speakers are in general. Basically, we have a wire, and when, say, a microphone vibrates this wire, it creates an electromagnetic equivalent within that wire. This signal gets transmitted into the speaker, where an amplifier brings up the signal by a lot, which gets heard within the “chambers” of the speaker. Speakers consist of three parts: a box, a crossover network, and diaphragms, including the voice coil, a necessary part of the diaphragm. A box is the cabinet that holds both the crossover network and diaphragms, which in fact are what is traditionally known as “speakers.” This is usually a subwoofer, for low end, and a tweeter, for high end. The crossover network is a device that is needed so as to not do damage to the tweeter and subwoofer. The crossover network divides the incoming signal into high frequency and low frequency portions that get sent to the diaphragms: to the tweeter, for high frequency sounds, and to the subwoofer, for low frequency sounds. The diaphragms are the devices traditionally called tweeters and subwoofers, but they are called diaphragms in the audio industry. The most significant part part of the diaphragm is called the voice coil, which is the electromagnetic signal that enables the diaphragm to act as a transducer, or a device that turns one energy into another. Speakers take electromagnetic energy and turn it into sound energy. It does this with the voice coil, which is an electromagnetic signal that interfaces with an opposing, stationary magnetic force, which makes the diaphragms vibrate, producing sound. This vibrates and moves the “speakers,” or diaphragms.

Microphones

Going over speakers was a bit daunting, but after learning that, it makes learning about mics a whole lot easier. A mic is just an inverted speaker. In fact, you can use a speaker as a mic if you wanted to. Microphones work in a very simple way, as I was saying. Sound vibrates a diaphragm which is connected to an electromagnetic coil and wire. The vibrations produce an electromagnetic equivalent that gets sent to an output source, usually speakers, such as in a PA system.

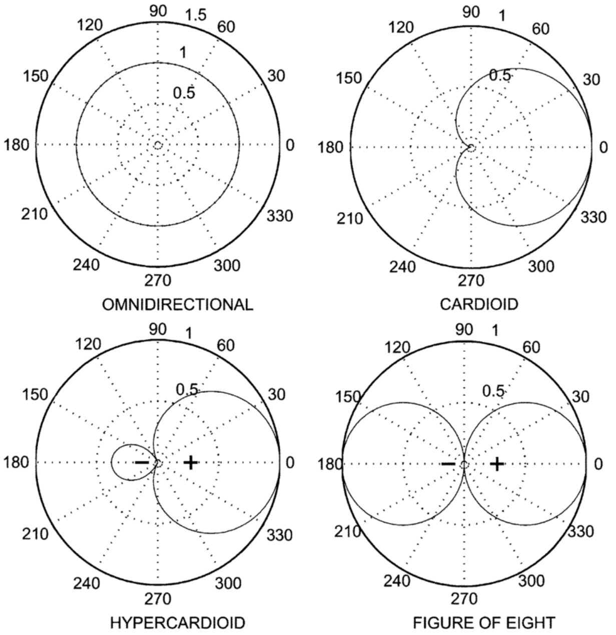


Condenser microphone.

There are generally three types of microphones we will be going over: the dynamic mic, condenser mic, and pencil mic. First, the dynamic mic is the most used as it is a microphone that does not require an external power source. All the needed electronics are fully integrated with the microphone. They are usually able to withstand the loudest noises and are used often for drums or vocals. Condenser microphones have more sophisticated electronics and require "phantom power," which is DC power, in order to power the more advanced circuitry. They are used for studio recording often, or in live studio performances, among many other things. The microphone enables sensitivity that can capture very subtle sounds and capture all the details of different types of instruments. They are usually a lot larger and come with different types of polar patterns, which I will be explaining in a short bit. The last type of mic is the pencil kind, which is traditionally a condenser mic, but is a small version of the condenser, and is usually used with string type instruments as the microphone can fit in small spaces and record very closely to a guitar sound hole, or piano soundboard. It picks up similar to a condenser, and the differences are quite subtle.

Polar Patterns

There is a lot to learn about polar patterns, but I will be certainly go over the good stuff, hopefully without getting too complicated. To begin, a microphone can be said to have a 360° circumference, with the front face at 0°. The idea is that polar patterns enable better sound positioning at certain angles. A cardioid polar pattern, or "pickup pattern," has a better pickup at 0° than from the sides or the back, where the back is expected to be least sensitive. The frequency response is expected to be best at the front, basically. Supercardiod is similar in that it has a similar construction, but with more 'side rejection' and less 'back rejection,' meaning sound is better at the back than the sides. Cardioid type microphones are the most used since for most recording purposes, front face sensitivity is most useful. You might see where we're getting at here. In an omnidirectional pattern, there is no rejection, and sound is expected to have similar sensitivity on all sides. This is similar to figure 8, which has the least sensitivity from the sides at 90° and 270°, and the most sensitivity at the front and back at 0° and 180°. This is good for sounds that might be bleeding in from the sides. Polar patterns are a necessary focus when dealing with microphones, and before we finish, there are a couple more terms to understand.


90
1.5
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0
210
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240
300
270
OMNIDIRECTIONAL
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1
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150
0.5
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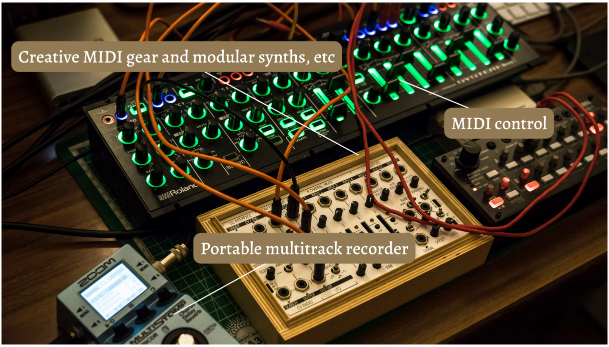
FIGURE OF EIGHT

Main polar patterns.

High end microphones are everywhere, so we will need a bit more terminology. 'Distance factor' is a way of determining the ability of any of these pickups to preserve the ratio of direct to reflected sound in comparison with an omnidirectional pattern: basically how does it stand up to omnidirectional mics. This enables a user to determine which microphone to buy and use at which distances. Another term is the 'acceptance angle,' which is used as a way to tell users to not place sound sources away from pickup angles that are reduced by 3 dB's. One of these angles would be at maximum reduction, which is the angle of least sensitivity. Or if rear or side rejection is less than 3 dB for instance.

MIDI

MIDI stands for Musical Instrument Digital Interface, and that's exactly what it is, an interface for instruments, whether they be giant Wurlitzer organs, or simple MIDI controllers. It is a digital interfacing scheme that enable notes, pressures, and duration to be recorded as MIDI notes. It is a very useful technology in the home studio recording world, but it is used everywhere, from television music production to advanced music machines.


Creative MIDI gear and modular synths, etc
MIDI control
Portable multitrack recorder

Typical "MIDI musician" setup.

In how we are trying to use it is in the world of basic computer MIDI technology, which usually involves a USB powered MIDI controller, which is usually a keyboard, drum pad, or MIDI mixer that connects to the computer. These signals are inputted to your DAW via MIDI routing schemes that enable MIDI responses within your plugins and DAW buttons and sliders. Configuration is usually the most difficult part in getting up and running with MIDI, so this is what we'll talk about next.

In every DAW or sequencer, and even MIDI accepting technology, there will be an option for configuring controllers, which will be your connected MIDI device. It is usually as easy as selecting the right controller, and enabling it, and checking to see if corresponding notes, buttons, and faders are activating upon operating the MIDI device. At this point, recording MIDI is enabled, and you can now record MIDI data into your session. The most likely use will be for recording notes and drum data onto a piano roll, which are corresponding notes, duration, and pressures to create MIDI compositions. The great thing about MIDI is that different patches, or instruments, can be tested with your MIDI notes to find good matching instruments. With drum patterns, this is the same process, however with percussion sounds. The same idea applies with testing drum sounds, but also quantization, which is a MIDI concept that enables syncing of MIDI data to various styles of musical timings and swings. These are all important concepts, and we have brushed on them a little, but entire books are written on this topic alone. Much can be done with simple experimentation, so I suggest playing with all the different areas of MIDI and find the things that are useful about it.

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Recording

Chapter 2

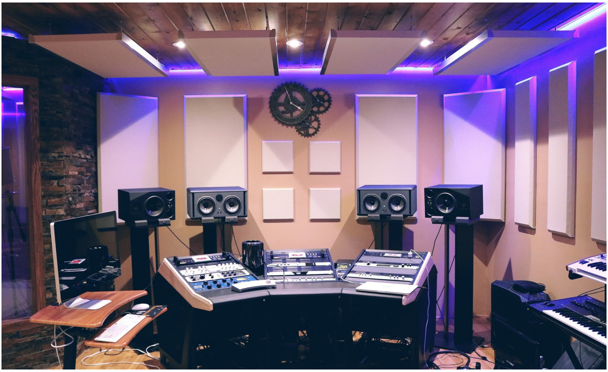
Recording

Multitrack Recording

We have gone through a lot of material, but now we will want to focus on the actual process of recording. A lot goes into good tracks, so here I want to distill the basic processes and functions needed to get started recording quality audio.

Multitrack recording has gone through a lot, and there is a lot of history behind it. Since the days of Thomas Edison, a 20th century inventor of the phonograph, we have innovated and come a long way. The phonograph is the first stop to understanding all this, invented in 1877, it allowed the vibration of a diaphragm to control a recording stylus on tin and wax. The recordings were crude and it was difficult to reproduce sounds in high quality, and so thus was a bit of a novelty during those times. But this allowed companies like Western Electric in the 1920's to invent more sophisticated recording technologies. This time, engineer Orlando Marsh was able to use a microphone amplified by vacuum tubes to drive an electromechanical recording head. This enabled the first recordings that caught a better audio reproduction that represented the original sounds and songs better.

Reel to reel is where we will go to next, as this paved the way for true commercial recording. During the 20's and 30's, much magnetic recording was being used during the war, by both Germans and the U.S Army. Different biasing techniques, or AC/DC additions to the audio, were used to enhance the quality of recordings, but it wasn't until Jack Mullin of the U.S Signal Corps that reel to reel recording became a commercial possibility. Jack Mullin brought his ideas to singer and actor Bing Crosby in 1947 with the intention of commercializing his machines. Bing Crosby invested $50,000 into Ampex so Mullin could help them with this new technology, which led to the introduction of Ampex's popular line of recording products, including the first commercially successful multitrack recorders. An Ampex 200 was given to Les Paul in 1948, further popularizing the new technology. Ray Dolby, a notable audio engineer, known for his high end sound technologies, was on the Ampex staff at the time. With his work on Crosby's machines, this led to the first videotape recorders, which was used to record Crosby's shows. This was the introduction of modern electronic media.



State of the art recording studio.

Even today, reel to reel recording is still highly regarded as more natural sounding to many artists. It can be argued, but many agree that the natural 'bass sounds' from harmonic distortions on tape and the natural compression in the mid to high range is more pleasing to the human ear. Nonetheless, it was this medium that took us into the era of the compact cassette, first created by Phillips, then the DAT, or Digital Audio Tape, and finally hard disks. Until the 1980's, recording studios used Crosby and Mullin's technology. When digital audio was introduced with DAT's, a shift in the recording world occurred, surely, but not without resistance.

Here we are now to deal officially with recording ourselves, and this is no small task; but it's not an impossible one. We just need to get started, and keep moving. Before we dive in fully, I want to mention the importance of 'default settings' in recording; and, yes, in all seriousness. The default options are going to be ok, but knowing what is going on is very important. It is like looking at chess players, and the setup. The chess pieces are there, and the opening moves are easy enough, but the further in you go, the more demanding everything becomes. Become good at using the default template, and recognize there is no magic going on, just the necessary functions setup for you to begin recording. I cannot express enough the importance of saving your sessions, and using your sessions as templates for future projects. Sometimes computers crash. This can destroy your day, and might even be devastating. Do whatever it takes. Backup your files if you need to with a cloud storage company, or use an external hard drive. Beyond this, be sure to create a restore point on your operating system to recover your computer. And last but not least, be sure your system specs can run the programs you wish to run. If I could change one thing about my beginnings in audio, it would be to use faster computers. What this does is it opens up worlds of possibilities. No longer are your plugins choppy or your sequencer glitchy. It all just works out.

Waves

Waves are probably obvious enough since we see them all the time in all their various forms, but a problem arises in trying to equate them all as the same thing. Sound waves are longitudinal waves, as opposed to things like light waves, which is a transverse wave, where the wave moves up and down through time. In a DAW, sound does indeed take the transverse wave form, but in the physical world, the wave is a longitudinal wave that is distinguished by its compressions and rarefactions as it propagates. Rarefactions are the less dense areas of sound as it propagates, whereas the compressions are the more dense areas in the sound wave as it propagates. In transverse waves, these are translated into peaks and troughs representing the compressions and rarefactions from the original longitudinal wave. In a DAW, they are seen as low amplitude, meaning low sound volume, and high amplitude, or high sound volume, in time.

DAW Settings

DAW Settings are very important upon learning how to use one, as there are many common mistakes, as well as many parameters that need to be worked with before one can begin using the software. Audio settings were briefly undertaken, and ASIO driver setup, but there are usually many more options, such as plugin options, whether they are VST, DX, AU, and options for choosing your samplerate and bitdepth, which we went over in detail. Any external plugins need to be searched for within your computer's system file, but there is most often an option for scanning for these plugins. You must provide a root folder so the software can scan for the program files to run within the DAW.

Samplerate, bitdepth, and other similar settings should not pose a problem, since the default options are going to allow for basic high quality audio input: 16 bit at 44.1. There is options to record at lower bitdepths, and even lower samplerates, but the convention is recording at 24 bit or 32 bit float, which, like mentioned before, represents the resolution of the amplitude in digital audio. Specifically, 16 bit in audio means that 16 binary digits, 1's and 0's, combine as a 16 bit word that allows for 65,536 possible levels. 24 bit allows for much more "headroom" by allowing for 16,777,215 possible levels, and at 32 bits, 1.4563 x 10^6 possible levels, which is an incredibly large number for resolution purposes. 32 bit float represent a dynamic range of 1528 dB, which is far beyond the difference in sound pressure that can be created on earth, which is 210 dB in an anechoic chamber.

Bits per second in a wave file is referring to the number of bits needed per second of digital audio, which is as simple as taking the bitdepth, such as 16, and the samplerate, say, 48 kHz, and multiplying the two: 16 x 48,000 = 768,000 bits per second. The same goes for 24 bit, or any scenario: 24 x 48,000 = 1,152,000 bits per second per 24 bit, 48 kHz file.

After all this, just to be clear, most professional studios record at 24 bit, 48 kHz, since it allows for plenty headroom, and goes beyond 16 bits in resolution. 24 bit is fixed point, also 16 bit audio, but 32bit is floating point, meaning, in the binary world, a word is not created out of the binary numbers. The logic for the computer is in scientific notation. This is the product of 32 bit audio files; it's a bit different. It is often much more than is needed, but does allow sound designers to experiment with loud and 'sudden sounds,' among other things. Further, 0 dBFS is the largest level of audio, which is communicated to the DAW. This is to situate this logical number per audio file. This is because 32 bit float comes with 770 dB beyond 0 dBFS, which is +770 dBFS, so it needs to be brought down, as it will come in clipped. It must be attenuated to 0 dBFS or the DAW cannot work with the sound correctly.

Tracks

When recording audio, tracks are used to record the audio. Each track has within it the ability for basic tasks, such as choosing input, or routing, mute, solo, and record arming.

Mute is your traditional mute used in computer applications; it silences the track until the button is deselected. The same goes for solo. Solo mutes all tracks except the soloed track for audition purposes. Then for recording, record arming is required to record in that particular track, so when the record button is pressed in the DAW, the audio will be recorded on the armed track/tracks only.

Tracks can contain multiple audio files along the timeline, usually seen in a "view" within the DAW. Tracks can also contain MIDI data for playing MIDI instruments or recordings. Envelopes are often used, which contain MIDI data, for different effects. Fade in's and fade out's can be done with envelopes, as well as tons more things. Tracks are simple components and should be considered simple, as over complicating tracks is antithetical to good engineering, since tracks are where you express creativity. They should be no more complicated than the functions I just mentioned.

DAW mixing

Mixing within the DAW is the most applied approach to recording and working with audio. A track is recorded, then applied a mixer channel, and within this mixer channel, for instance, in "channel 02," an effect plugin is often applied, be it a VST or default plugin, such as a compressor module. The audio is then processed by the plugin, resulting in a compressed version. Let's say this was a guitar part, and recorded as a three minute instrumental, and you put compression on it. This is a starting point. The next thing to do would be to either cut off unnecessary audio in the beginning and end, or just crop the track. For more precise viewing, you can change views into an editing view, and do more precise cuts. There likely will be areas of silence with background noises that need to be silenced. And in no time at all, there should be there sitting a clean guitar track. The next thing to figure out is, does the track need more fixing? Does it need eq? How about reverb? We then add an eq plugin and reverb plugin to the same channel, make some adjustments, and voila! It can be this easy, and if the guitar was captured perfectly, then it usually is this way. Not much else should be done. This is a good starting point. Especially if it is a podcast or a vocal recording, it should be kept simple like this. More can be added, such as bass guitar, drums, and vocals. Even these recordings require very little, and should be left clear and untouched unless totally necessary. Hiss and hum can be a problem, but there are differing theories in the industry, and I do not want to shake anything up, but I would simply suggest using it if your situation calls for it. It is about your room and studio environment. Not everyone has a recording booth with perfect acoustics, thus it becomes a situation to learn about. Traditionally, unwanted sound is sampled and is used as an example of what to remove within the entire audio file. Sometimes this can damage the audio. If the unwanted sound isn't too loud, then it could be a good option. I would recommend trying it if you can get enough acoustical control to quiet the environment. If your acoustics is under control, there shouldn't be any problems with hum or hiss. It can be a mistake in the long run. We will be explaining more about acoustics later.

Plugin FX

Like mentioned, our guitar with compression is really good, especially if cropped and edited well. It's a take! This is easy enough, and it should be, but sometimes you will need to go in and make more adjustments. Traditional plugins I am going to talk about are compression, eq, and reverb. This is the basis of all great recordings, and often eq and reverb isn't even needed.

If the compressed guitar had a ratio of 2:1, that is a great thing. That means that the peaks in the guitar recording didn't need much compression to "level out" the recording. Now we can apply some eq. We throw in a 5 band parametric and decrease the high end around 5 kHz, to bring up low end (more on this later), and decrease the low end around 100 Hz (to bring up mid range and high end), and there you go. The science is all in equalizing for a more natural outcome. Using your ears is not just a saying. It is as simple as recognizing at which frequency level there requires an increase or decrease. These sliders, or knobs, that indicate the frequency band are much like volume controls. You would be "boosting" frequencies in this range, or making them louder. Or making them less loud of course. Good. Lastly, a basic reverb plugin is thrown in the mix, which is just an ‘environmental emulation.’ It is a room, where you can choose the size: the height, depth, and width. This creates reflections in considering the different room sizes you choose. It can create things like warmth, which is why it is a traditional go to effect, especially for vocals and guitars. Also, the room can create ambiance that indicates space, and can create character in your recordings. The options on reverb units can be many, and can have embedded effect units, such as eq, but, mostly, the knob we’re looking for is the ‘decay.’ This is the time it takes for the reverb to die away, and is wet the more reverb it has, or decay. Pre delay can help with this idea, since more delay helps separate “dry” audio from the reverb with a short delay. Pre delay can help situate where you want the audio to sit in the mix, since the more wet, or effected, the audio, the more it sits in the background.

We added some mid sized reverb to the guitar, with just a tad bit of wetness, from decay, and a short pre delay, which makes for really natural sounding guitar. That’s it! We can play back the recording, and even mix it down if we want. It should sound great.

Instrument Plugin

Let’s take our guitar and add a VSTi plugin into our track space, or view, and record some bass guitar on our MIDI keyboard. It’s as easy as choosing the “patch,” or instrument preset, and arming the track for recording MIDI. Use the option for a count in, or 4 count beat, in your DAW and hit record. The great thing about MIDI is that you can edit MIDI, so if you mess up, you can go back and change the notes in the piano roll. There is also options for ‘quantization’ which are ‘pre alignments’ in considering traditional music timings and styles. Don’t get too confused, though. We recorded bass for the first verse, and now we can edit it how we wish, and then copy it to our other verses in the guitar song. Here we choose to play bass a little bit different for the chorus, so we arm our MIDI instrument and hit record. Boom, we have a new chorus part! If all has worked out, we have a pretty good thing going. Now we can add a bit of side chain compression. Play it back, and it should sound good. There is always the option to experiment with different patches, equalization, and different effects. Comparing your recordings early on is a great way to get your music sounding good. Just A/B your song against another song. Meaning, mute it, then solo your song. Then do the same for the other song, and you can get a good sense of the frequency response, vocal wetness/dryness, drum reverb, etc. Now let's add some drums. The track is getting groovy.

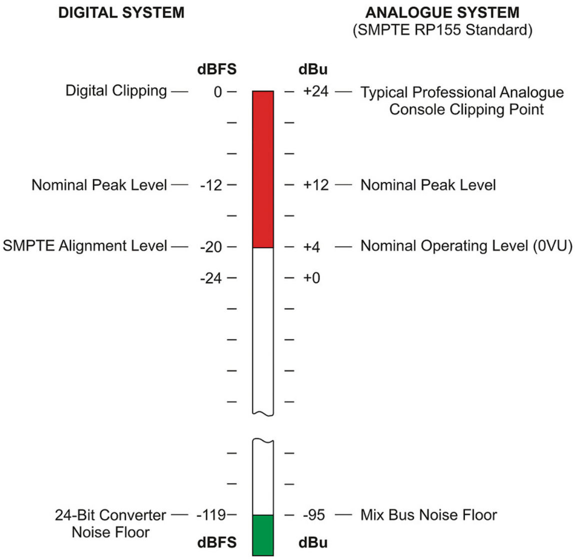
Noise Reduction

Noise reduction, like I mentioned, is a serious thing, since it can damage your tracks, as fidelity is lost almost always as a result of using these technologies. Good sounding rooms are important, and doing it right from the beginning can save a lot of time. It is an option if you wish, and it is different depending on the DAW or plugin. Most of the time it consists of taking a sample of the constant hiss noise that is creating problems, then applying this to the whole track to remove this hiss in general. What often happens is it alters the quality of audio, though, thus it is best used in ways that are unnoticeable.

It looks like we have a slight problem on our drums. It was recorded in another room we put the drums in, and gosh there is a slight hiss that is ruining the track! We sample the sound, apply the sample, and boom, we have a clean drum track! A little compression and reverb, and voila! Drums are one of those things, too, that have very little to do with effects, and much more to do with your recording equipment, microphones, and initial capture. Playing well and capturing well is mostly what is going on here.

Unity Gain

Unity gain is not so obvious for beginning students, and this is not a good thing. Unity, or ‘gain staging,’ was the last thing I looked at when I began recording. And I was using a 2000.00 dollar machine at fifteen years old! All I knew is what sounded good. There were lines that indicated unity, but I relied on anything above the 0 line to be relatively useful. I relied on this way of dealing with audio even after getting into DAW’s. It is a must that I come forth and give some reasoning to this situation. In the digital audio arena, anything above the zero level on your faders is a no man’s land, so mixing above unity is not a very good idea, since digital audio is not meant to be played above unity as distortion will occur. Mixing at low levels is how you fix distortion issues in your mixes, or add gain reduction plugins on all your mixer channels. Nonetheless, a habit must be made in working below unity, and amplifying sounds formally during gain staging.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ANALOGUE SYSTEM
(SMPTE RP155 Standard)
dBFS
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Digital Clipping
0
+24
Typical Professional Analogue Console Clipping Point
Nominal Peak Level
-12
+12
Nominal Peak Level
SMPTE Alignment Level
-20
+4
Nominal Operating Level (0VU)
-24
+0
24-Bit Converter Noise Floor
-119
-95
Mix Bus Noise Floor
dBFS
dBu

Unity gain on digital vs analog systems.

Everyone in the modern audio industry says to "mix loud," and all this means is mixing at or just below unity. My experience says that good tracks come from good equipment, instruments, and plugins, and playing well, and sure, at unity, but this basically is more the way of making big and full sounding tracks.

Pre and Post Production

I don't want to scare anyone off as if there is some magical pre and post programming thing that must be understood in order to record, but it is a good thing to note the whole idea of pre and post production. All this means is work before mixdown into a tangible wav file, such as creating an instrumental to bring to another location to record a vocalist, or premastering edits, such as precompression before the official master edit. Post production can be recording extra parts, overdubbing in various ways, mastering, sampling up or down, or even filtering the song through old reel to reel equipment or outboard gear. There is a lot that "could" go on in post production. It is about who is involved, and the process chosen to finish a track. It is good to be well versed in this area, since upon working with producers, engineers, artists, companies, etc, they might require certain preconditions, or do mostly post production work, etc.

Panning

Panning has come a long way since the early days of recording and many emergent ways of panning have become accepted in different genres of music. What I want to do here is go over what it is, how it's used, and some templates you can use for your recordings.

Typical rock band stage setup.

Panning is a stereo effect, so there must be a left and right channel, or left and right monitor. At center, this is equivalent to mono, or single channel audio. In this setup, there is no way to obtain a stereo effect such as panning. Most modern audio equipment have the ability to play sound in stereo, as there is always a left and right channel in audio file formats. This is where panning comes in. Panning begins at zero, and left to 100% and right to 100% in general. This situates the audio more toward the right field or left field in the 'stereo image,' enabling stereo sound. This does wonders with certain sounds, such as bass guitars, drum sounds, vocals, and more. Particularly with vocals, a boring vocal with no panning can be turned into two layers of overdubbing both panned equally left and right 50%. This creates a full stereo effect and can bring excitement into your recordings. This is just the basics of what you can do, and in panning, a lot of it is about creativity, but also balance, since panning can be an overused thing, too. That's the secret to getting good at audio; using your talent and ear to use effects and audio ideas at the right time with the right things.

Rock music is held high in the pan world as it is a good template to understand panning in general as it is basically what is more real and appropriate; their stage locations I am saying. Dead center you will have your bass, kick, snare, and vocals, and left and right for guitars and keys, with a good spread for overheads. Center is a good place for most things, and often times panning is a very simple thing for many audio professionals. Hard left or hard right is the case for many in the industry. This does not mean there aren't exceptions for them or others. Center is good for drums, bass, and vocals in general, then off center will be things like synths and guitars. This is the convention, but there are some things to consider. The pan law states that two identical signals panned hard left and hard right will produce

6 dB's when combined. This requires that if this is a technique you are using, you must compensate for this by decreasing by 3 dB, or 4.5 dB as the effect could be dramatic, especially in settings where only mono is available. Here, the effect will just be a high dB increase!

Overdubbing

Overdubbing, also known as layering, is one of the critical components of sound recording that enabled very quality recordings in recent times. This went about not just by tracking different parts, it also did this creatively by layering vocals to make it sound like more than one person was singing. Chorus's became more interesting, and many things became possible. Traditionally, singers will choose certain parts to be overdubbed, and often they will overdub the whole chorus, sometimes adding harmonies. In other areas of music, such as experimental music, or pop music, overdubbing can be used for recording scenes, background skits, intro speeches, and the list goes on. This is especially important in film audio where sometimes sounds need to be added in, vocals overdubbed, such as for narration, and for foley, which is industry speak for creating sounds, usually natural sounds, for film. Overdubbing is how I started on tapedecks when I was only thirteen.

I used to record myself over instrumental music, experimenting with layering. You would use two tapes, and after creating one take, you use the other tape to layer and record the audio from the other tape. It was great fun. You should try it. You can also use this technique to create an old-school effect.

Traditionally, songs in pop music are more ‘changing’ than older music, such as early chant music from the dark ages, a period of “barbarism” if you wish, that kept a rather unchanging, single note progression, and orchestral music that kept a very simple harmony. Modern call and response became the norm, much of which due to the improvisational potential of certain styles. Here, lines one and two were called and repeated to give time to think up the last response line, which was often a punch line. There is much to say about this as a lot of modern pop stems from early blues and jazz styles. In modern times, songs rely on a similar technique applied to verse and chorus, or chorus and bridge, which is a call and response structure. Verse, chorus, verse, chorus, verse, chorus is the basic scheme we all know in modern times, but this can be turned into many different song structures. The last verse can be turned into a bridge. Or a fading, extended chorus can be added to the end of the song. The idea is simple; you call up a verse, and respond with a chorus.

Lengths can vary for versus and chorus's, but 12 bar verses are normal for modern pop songs, with 8 bar chorus's. A bar is 4 counts, so a verse is 4 counts 12 times while a chorus is 4 counts 8 times. This is assuming a 4/4 time signature with a medium tempo between 70 and 100. The faster the song, the more space you need for a verse or chorus. The less fast the song, the less space you need to create a song with a traditional length.

Mastering

Mastering is the eventuation of a good and complete recording session. After mixing down the song, the mastering process should start. Or it can happen after other post production stages, but after a good, solid, well mixed outcome.

Mastering is the art of commercializing the audio. This will be about a number of factors: sounds of other, related mastered material, the production sound level of similar commercial recordings, the tradition of certain genres or industries, audience, and more. What is needed should not be much. If what is desired is not already accomplished, mastering is not going to save the recording. This stage is for appropriating things right so the outcome is more ‘streamlined.’ It’s more like good management.

iZOTOPE
OZONE 9

Izotope Ozone mastering suite.

Some factors that can be important are often low end rumble, distortion, high end issues, and monitoring issues. Cutting the low end at around 80 Hz is common in modern audio production, as this low end can pose problems on the subwoofers of many monitors. Getting rid of this clears up the sound a bunch, and the useful low end is presented well. Distortion problems might be an issue, and warming it up, decreasing harshness and high end frequencies can be done here. Testing the sound on different monitors is good practice because distortion might only be a problem on some monitors, and not in others, so getting a feel for the situation can go a long way. Anyway, if the mixing was done right, below unity, things should be ok.

This was just a taste of what mastering is about. We will be going over more in a later chapter.

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Acoustics

Chapter 3

Acoustics

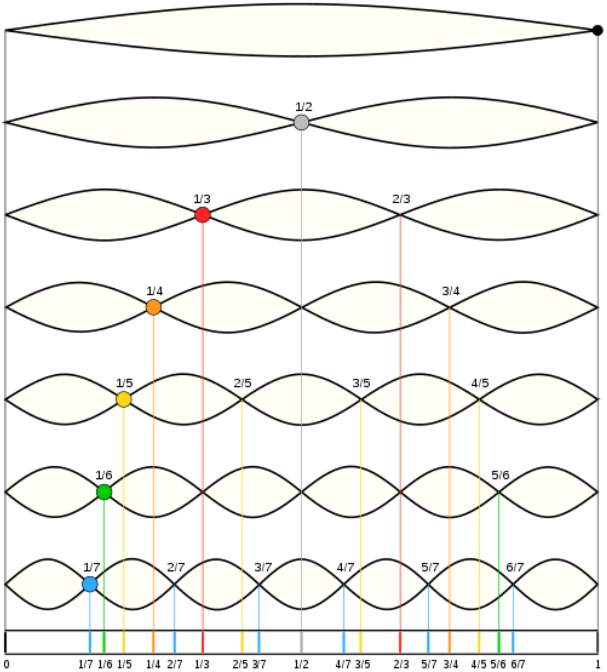
Tone

Tone is about how the sound "sounds." This is related to harmonics, and we will see soon what that means. Different instruments give different tones, and even the physical space contributes to the tone of the sound being produced. The fretting and slides of stringed instruments, the embouchure, or mouthpiece, of brass and woodwind instruments, or the mallets in percussion can create different tones.

The perceived color of sounds with differing tones is the 'timbre,' which helps distinguish sounds from entirely different sounds, such as a bicycle from a motorcycle, or a guitar from a ukulele. Timbre helps us create textures in the audio production world, such as when orchestras, jazz bands, or rock bands come together. Especially for singers this is useful, since some people sing higher than others:

soprano for females, or counter tenor for males. This differentiates pitch, a higher or lower octave, so we can create harmonies that create an interesting mix of timbres. Other options are: messo soprano, mid range, and contralto, low range for women, and tenor, high mid range, baritone, low mid range, and bass, the low range for men.

Harmonics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Harmonic series in acoustics.

Harmonics, like I mentioned, is related to tone. This is because the tone perceived is created considering the harmonics involved, and harmonics are different for different sounds. Harmonics are usually identified by their initial pitch, then stemming from that create a second harmonic, a third harmonic, etcetera. These are what set apart different instruments, since without harmonics, most instruments would sound boring. These harmonics are natural outcomes of how the instruments work. These are also called 'overtones,' and these overtones can be very obvious in certain playing styles, especially in guitar music, where higher harmonics can be heard when playing certain chords.

Room Acoustics

It has been a long time since I was first interested in room acoustics, and no one knows how far I went, but it is a common cause among all audiophiles. The room can even be portioned off with egg cartons, or mattresses, foam, curtains, cushions, pillows, blankets, you name it. Where there's a will, there's a way. The science is not a science at all in a way, unless you want to get into the calculus. Most of that is unnecessary, but that does not mean we should throw away the basic ideas.

Diffusion is the idea that sound reflects off of a surface and does not reflect perfectly; rather it diffuses into surrounding material, and bounces in different directions, creating even some room reverberation. There are many ways you can create, or hinder, diffusion. Too much material can lead to too much diffusion, and this is probably not what we want. We want the sound of the room to play a part in our recordings. Diffusion often is added to a room by adding acoustic foam in various places of varying sizes to enable a good balance of reflection and absorption. Diffusion is simply the scattering of sound energy and used by many acoustically treated rooms to create a reflection outcome appropriate for the venue; hopefully with sound perceived the same throughout the space. There is much going on in this area and much to check out. One idea that is a must to understand is that, sound treatment and soundproofing are two different things. Soundproofing is not necessarily what we want unless we wish for a very dry sound for adding makeup later. Usually what we desire is sound treatment to add a certain character to the room. Dampening is the next important idea to understand in all this. Dampening is what we do when we want to reduce the sound reflections in a space. This is what many sound absorbers aim to achieve, since too much reflection can cause unwanted room characteristics in certain areas. Many materials can help with simple dampening, such as book shelves, blankets and carpets, sofa's for artists, among other things.

The last thing I will mention briefly are bass traps. It should be noted that what we're referring to by the sound of the room is room for listening, mostly. Monitoring music in a good room is just as important as creating music and recording in a good room. Bass traps are important for obvious reasons. They are important for the low end in using instruments, singing range, etc, but it is also useful in mixing music. These help with controlling the low end, and as you'll see in the next section, they can provide a good setup for our listening environment.

Professionally treated room.

Acoustic Treatment

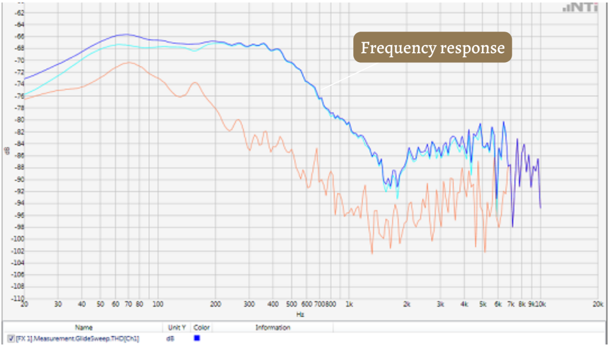
I want to give a very general idea of this so you can leave this and have a basic grasp of how this works. Simply, in a mixing situation, the proper speaker (monitor) placement is included in the manual. However, it will most like say to situate the monitors at ear height, and this should be good, probably rotated slightly by 30°. Beyond this, there does occur reflection problems, since sound travels one foot per millisecond, thus some waves will reflect off of certain surfaces before reaching your ear. This is where sound treatment comes in. It will be a good thing to do this, since without it, you will not be hearing an honest version of the audio in the room while mixing. Directly to the right and left of your mixing position, hopefully near the center of the room, at tweeter hieght, treatment should help absorb these reflections, because these are immediate reflections that will add to the sound coming out of the monitors. Directly behind the speakers can also pose reflection problems, thus treatment here will go a long way. Directly above the speakers is also a very important location for treatment. I think you see where I'm getting at with this. It is your listening location and perceived reflections that matter when mixing in a room. In a recording scenario, the problems are a bit different, but essentially, you want to control the reflections so the right amount of reflections reach the microphone through diffusion and absorption. Bass trapping is another thing to keep in mind, since this often times produces an overall good sounding room with "cinematic quality." This occurs because the overall bass sounds are more controlled and come off as more pronounced, and "even." This helps overall in music production and recording, especially in listening. Boomy rooms can occur because of materials, monitors, room size, etc, and so bass traps can help prevent this boomy phenomena and even out the sound.

It is a rather large field and entire books are written about this. If you'd like to get into the physics more, I encourage you to explore.

Microphone Placement

The hard parts of audio production sometimes can come from interesting parts of the audio industry. We've all heard of 'mic placement,' and heard of how early engineers used to record. It was very simple in those days. The importance is still there in my opinion, but there are certainly some more modern things you will want to understand. One of these things is the type of sound being recorded. The sound hole from a guitar along with its overall construction is different from a vocal recording. A common way in which to achieve generally good results is through 'measurement.' Measuring the height of the 'sound producer' can give a good distance to use between the microphone and the sound producer, such as a guitar, which is around 38 inches. Mic placement 38 inches from the guitar is generally a smart decision. There are other ways of doing this, such as placing the mic near a source, such as near a sound hole, sound board, or fretting, etc. This produces varying results, since not all of the timbre from the instrument comes from these particular places. However, placement around these areas can have unique and interesting effects. The last option I will mention is using a PZM. This is a Pressure Zone Microphone used for boundary recording, or hemispherical recording. It is used to pick up full room sounds, and consists of two microphones placed perpendicular on two walls, enabling full room capture of sound from their respective boundaries. Capturing a good room can be done well in this way and is a great way to experiment with more advanced areas of audio.

Frequency Analysis



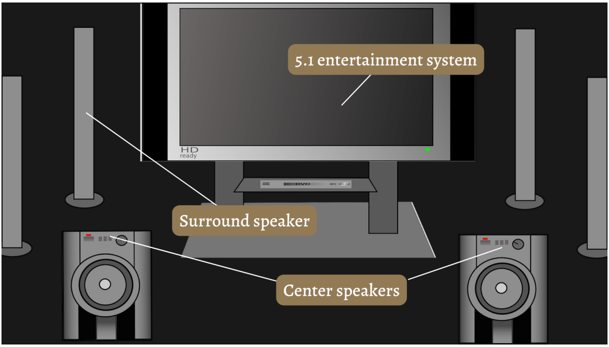
Frequency analysis graph.

In continuing, the end part of the chapter I want to talk a little bit about is frequency response. It is a catch all term in the audio industry for the response you get from microphones, speakers, audio processors, in DAW's, and in rooms. Since we are on the topic of room acoustics, I want to give a basic explanation of how frequency analysis can help your recordings. Frequency analyzers are old audio technologies created for the purpose of assessing the frequency response of different things. It's usually as simple as setting up a mic, speakers, or playing some sort of sound, and running it through the analyzer, hopefully getting back a graph. This enables you to see where you need to remove unwanted frequencies. Normally, you want a flat frequency response in a room with no issues in the low, mid, or high end.

If there are major problems, say, in the low end, cutting frequencies with bass traps can go a long way. Cutting the high end also works the same way. If there does exist big problems here, then using treatment can fix these issues. Frequency response and analysis can be used in many areas of audio, and this is definitely a good start in getting to grips with all this.

Surround Sound

Mixing in surround sound has been an option for some time, but it is all more complicated than just starting to mix like this. The general view is that you mix with many speakers in mind, usually five, but with a left and right main, and sometimes a center. This enables using the various speakers to create spatial panning effects, cinematic low end effects, and other things. True surround sound is the idea of recording with 5.1 surround sound in mind for each speaker by assigning it its own microphone, thus making each speaker special in regard to how you setup your microphones. This is popular for film and cinema, but not much else. This is because surround sound equipment is needed, specifically a 5.1 speaker system, or a processor paired with multichannel speakers, and this is just not available without an entertainment type of system in mind.



5.1 surround sound system.

Another option is to include 3D sound in your mixes which are relatively equivalent to some of the effects you would produce in surround. Simply, speakers direct sound to our ears and reach them at slightly different times, which produces "cross talk," a phenomena that hinders our ability to distinguish stereo elements in a recording through speakers just facing us. Headphones solve the problem, but still it is not necessarily a perfect reproduction of how the sound occurs naturally. A better option would be to use 3D audio. Creating this effects can be done with specially built microphones, but an even easier route to getting started in this area is to learn more about Head Related Transfer Functions, or HRTF, by trying out some of the plugins out there. A popular one is the Binauralizer, and Binauralizer Studio. HRTF's describe how sounds bounce, scatter, and defract upon reaching the ear, and enables sounds to be inferred in regard to anatomically precise measurements. These functions allow for 3D audio that can be very interesting spatially since they simply boost and manipulate certain frequencies that are anatomically related. I would suggest checking out the scene and trying out some products to get a feel for this stuff.

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Production Secrets

Chapter 4

Production Secrets

Mastering

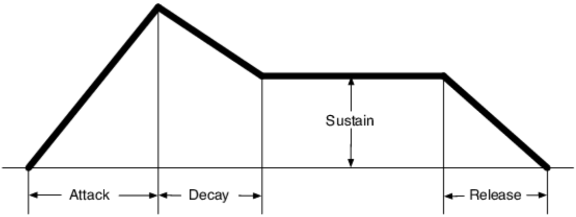
My ability to master is my own ears and ability, so thus mastering turns into an interesting thing. I am using my ears and my know how to create a finished product, and it might be very different from the next guys. Sure, we did that to create the overall product, but isn't this supposed to package it up? Yes, but there are many opinions out there about mastering. Some will tell you to try presets in Izotope. Others swear that mastering must be done by another set of ears in a grandiose studio. I take the road of least resistance and I balance the two. It depends on the requirements of the finished product. If it is for a personal recording, I might use a preset. However, if it is for something bigger, like a podcast or song, I might want to hand it off to someone.

Nonetheless, some principles can go a long way in dealing with mastering. Mastering has many layers, like an onion, and all layers must be dealt with in some way, even if it’s an ok, “no need to add anything here” type of scenario. Limiting, compression, eq are some of the tools we’re going to need. The first thing to consider would be limiting for reasons previously went over. This creates a ceiling so our mix doesn’t get out of hand. Once a balanced outcome is achieved, we begin to take corrective action, whatever that may be. Often corrective eq can help with certain parts that need work. This is where comparative mastering can help, or A/B mastering using a reference track as a desired outcome. Compression can help, too in this chain if needed, post corrective eq, and mastering eq at the end of the chain. Other methods can be used, especially multiband compression, but I would suggest keeping it simple and choosing the right technology for the right purposes. Another thing to keep in mind is making sure that you are not mixing at a higher bitdepth and samplerate then the original track. You want to keep the same settings, or downsample. Distortion can result with upsampling and this would be a mistake. This is because you would be adding unnecessary audio information to a signal recorded with less information.

My personal opinions are littered throughout, and here again I am going to throw a few things at you.

I have seen it all, and instruments should not at all be taken lightly. Whether it's a pop track, or a guitar jingle, there are formal ways of using instruments. The best way I would say is by letting the performer engage in creativity and formal playing, but sometimes the formality is not always there. Here we must create it. The problems that arise are often serious, such as playing off tempo, off key, or the vocals are a bit off or non traditional, or it's just "bad." I have come a long way in skewing things, and if you can prevent it, definitely do, by playing well, making sure that whatever is captured is as perfect as can be. Another option is looping instruments. All you would need to do is play on tempo for a few bars and loop it. Do this in different areas, and there you go, you are off to a good start. There are some things to note upon doing this though, and in this chapter, I am going to be going over this idea. In music, playing style is key. Maybe sometimes something is a bit off and funky, but usually the most off tempo parts are the vocals. Getting to grips with alignment, beats, and quantization is key in all this. Sometimes where to start is a problem, and if your creative juices aren't flowing already, I recommend starting with a drum track. We will get to drums soon, but for now, this can help with the playing. A click track will do fine, but elements can be a great help. Upon recording, be it a loop or a full three minute guitar track, it is a must that staying on tempo is understood. There are few things in life this important. It would be better understood with an example. Say we had a drum beat, and bass, and a drum fill 8 bars after the start of the song. Right at that moment, guitar comes in. It strikes with the power and force of the vibe of the song, but in a 'tempo synced' sort of way. It is best that this is understood early. Getting hooked is the job here, and these hooks come from good and accurate playing. No matter the case, any instrument should be utilized well, played traditionally, and played on tempo.

ADSR



ADSR plot.

This term is found all over the audio industry, and stands for ‘attack, decay, sustain, and release, and yes, it is a very general thing. It is found on synthesizers, audio editing programs, and various technologies, as it is an envelope which describes amplitude, frequencies, or pitch . It is what a waveform essentially is and is doing, too, like what happens when a piano key is struck. Attack is the beginning part of the envelope, decay is the transient, or highest peak and corresponding “fade,” which is attached to a sustained part of the process which results in a release of the process. This is essential knowledge since this is where editing can get pretty complicated. I went a long way without having to deal in this area too much, but it is handy knowledge for the long run, since it is seen everywhere, and it can help in sound design and envelope editing. This is explained here because I want to show how they can be used in using basic envelopes throughout the recording process.

Percussion

Drums are the base of most songs, and that is because it keeps a steady rhythm throughout. I have been in so many areas in music that it is worth really explaining this simply so it is digestible and actually useful. Kicks, snares, and hi hats should have no problem sitting in a mix right at the center, with some overhead cymbals panned to your desires, but one thing to remember is that this part sits in the middle of the mix often dry, thus wetness becomes a critical factor, including appropriate equalization.

Equalizing kicks is a pretty simple science, but it might not seem so obvious at first. A high pass filter, a filter that cuts low frequencies, at 50 Hz can help a lot to cut bad low end. In addition, boosting just above that can help with the kick sound itself, while attenuating frequencies between 100 and 250 Hz, since this could hog a lot of the kicks power. The snare should work with the kick and bass sounds, thus cutting at 80 Hz can help make sure that the snare does not interfere with the low end too much. Boosting at 7 kHz and 12 kHz can also help the snare's presence. Hi hats and cymbals also need to stay clean in the mix, and this is often done with high pass filters, too. They sit at around 500 Hz and 2000 kHz or higher, and they can greatly affect the low end, hence why it is necessary to manage these drum parts. I am a firm believer of natural sound, and I believe that good recordings come from understanding this. I do not believe unnecessary filtering needs to be everywhere. I believe that if drums need compression, it is for transient control, or peak control, and reverb can help create more wetness and color to the drums. Adding reverb can also do damage to drums. Room sound, if in good quantity, can hold up a lot better than digital reverb. Now I want to talk about a special process I have used a lot over the years, and this is about pitch, time 88 stretching, and offsetting. Changing the pitch of the track can be a lifesaver, especially if instruments aren't working well together. This enables the mixer to change the pitch to a higher or lower value that sounds natural in the mix. This is related to stretching, which is also one of those things that can help with single samples, but that need to be sped up or slowed down on the fly. Offsetting is a great tool, too when you want a sound to delay a little within the track. It can make a big difference, but it takes practice and familiarity with these controls. These are very basic editing functions and I suggest you learn how to use these sections in your software. Here we can change the pitch just a tad bit lower, and there you go, a near perfect drum track. Near perfect, because we need to learn a little bit about side chaining, and that is what we'll do next. Side Chain Compression What most people hear in pop songs that makes the sound so thick is many times attributed to sidechain compression. This is because it creates a fuller low end by using limiting on bass to change in considering another sound, such as a kick. This helps give room in the low end dynamically while the song is playing, and adds a lot of character.

First, you want to assign your bass track to its own channel. Then, add a compressor with an input for side chaining. Now select the chosen kick for the input. At this point, we must adjust the threshold and ratio to become compressed every time the kick hits. It is a widely used effect and can do a lot for a song. It is good in many scenarios and should be experimented with to get the best results. It works great on pop, hip hop, and R&B tracks, and can be used in many other genres, especially EDM and funk music.

Lastly, the next lesson applies to your drums, too, so pay attention. There is a lot you can do with the following ideas, and if you want good drum mixes, try to follow these ideas as much as possible.

Mixing

I want to touch on just a few things that helped me in learning to make more professional mixes, especially after failing a lot at this in the beginning. Gain staging is an arena that I encountered after the fact, and I want to pass a few notes. To many big pop and pro mixers, they rely more on this stuff I might say, but surely everything is a diversity, nonetheless. Gain staging is really just controlling gain for the end result, a loud, quality mix. You can lower gain in a mixer channel, within plugins, on you mixing board, on your guitar amp, anything. Or you can increase gain. The idea is to mix below unity, with the hope this results in a loud track. A kick, snare, percussion, even sounds that hit fast, as in a short attack, these create distortions that affect the overall track's capacity for loudness. In order to create high quality productions, this needs to be taken care of, hopefully early on in learning how to mix. All that is needed is to put a limiter on tracks that have this sudden loudness, or hit, that adds to loudness on layers within the recording, or song. Limiting this to atleast unity, or lower, greatly enhances the track's capacity to sound professional and loud. Engineers and producers all struggle, especially in modern times, with trying to cut these "db's" as much as possible, and this is also why high pass filtering is a huge deal in professional productions. If you can cut db's from any sound by altering it's low end, by using high pass filters, then this gives a lot of room to enhance the quality of your kick and bass sounds. We want good quality, we want perfection, thus the list goes on. It can be seen the myriad of ways of eq'ing, and you can be the judge as to whether a 70's track is high fidelity because vinyl or because eq, but I would suggest that whatever it is, it always sounds better eq'ing correctly. The primary reason most truly professional producers would say is the reason for why they eq the way they do is to take out high frequency distortions, in any track, whether it is vocal, guitars, etc, which enables the necessary mid range focus. You can see this in a lot of EDM. Like I mentioned earlier, to bring down the high end to bring up the low end: it is because it is always better to cut then to boost when trying to aim for a presence with eq, thus to give presence to high end, cut low end, and to give presence to low end, cut high end. This also helps in cutting db's for the overall necessity for loudness and quality mixing. It is a lot, and it can be pretty frustrating knowing what to do, but if you take it from me, it is a creative area. It is OK to work on it over time. It is either easy or not easy, and I would say it isn't easy if you aren't doing it, and if you are, you are probably still confused.

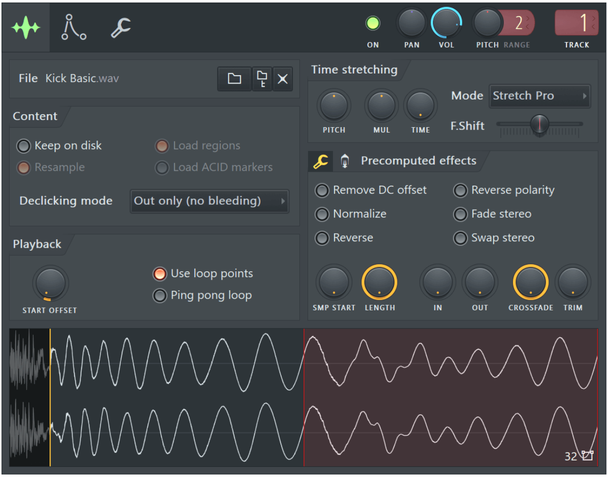
Environment

Every studio is an environment that engenders music and songwriters, and life, because this is where creativity thrives. Considering this, it is a place that is not only about recording and technology; it is a place for nurture and growth. Every studio environment should be tweaked to help artists 'breath easy,' since this sort of environment is quite nerve-racking in general. Not to mention, the temperature, shapes, colors, lighting, smells, etc all add to potential discomfort. All these variables should be realized and appropriated for all people within the environment. All things still apply, as well as how the business looks and operates. It 90 should not be a difficult thing to realize why this is all important.

Lyrics

Lyrics should "make sense." They should also flow well. There is no real science to it unless you want to point toward poetry or the arts. Songs are supposed to express emotion and interest. It should make you smile, grin, rock out, or bop your head. Political songs are great, too, since messages can be expressed emotionally, and this creates followers for powerful causes. It is one of the few mediums that can reach masses in such a short time. In considering the variables mentioned, lyrics and songwriting should be taken humanistically I might add. There is so much ideology, and religion, and culture, and belief, and politics that it can be difficult to know what to write about. Take it from the greats, such as Elvis Presley, and songs about being 'all shook up,' or The Beatles, who wrote songs about 'coming together.' The songs were often very simple with simple messages, pleasing to the ear, and accepted by almost everyone. It's easy to go in certain directions and get caught up in writing about abstruse things, but I would save abstruseness for the more personal, biographical elements of musicians if you want my opinion. In fact, the more formal you are, the more you will be accepted in music, thus, abstruseness should be learned about as a critical thing. No musician ever fully adopted a perfect musician mindset, but almost all left much vulgarity behind. Vulgarity is useful in music, but it is only a part of a larger whole in each genre within the music industry. Sure it is known for being a wild thing, but at the same time, adaption should be undertaken early on so as to accept all the more important theories about the music industry.

Timing



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Content
Keep on disk
Load regions
Resample
Load ACID markers
Precomputed effects
Remove DC offset
Reverse polarity
Normalize
Fade stereo
Reverse
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Use loop points
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START OFFSET
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32

Timing controls in music editing software.

This is where I want to get across something very important, and I want to express my thanks for you reading this far. It took me a very long time, 15+ years to get to a point of understanding music creation to where I could design anything musically. The secret was no secret, and really was behind what good musicians do already, and that is good music playing, or more simply, timing. Playing well is important, but more important is everything atleast sounding well no matter what it takes. I will give an example of what I mean by all this, but first: we need to come to terms with the professionalism of music. People are there to do good work. In professional arenas, almost no one does bad work, so you better know your stuff. Chord progressions will take you a little bit of the way, but it will only take you so far. Let's put four chords, three repetitive C chords for each 4/4 beat, and then one chord on G for the last beat. Sounds good. Now this is a basic chord progression. There is too much to know, but here I can show you some tricks to get by, and even take with you for songwriting purposes. See, the problem is context. There is no context in which to work from. A good way to deal with this is by live playing, or creating quantized MIDI tracks, where all of which should be seen as ghost notes. If this function is not available, there are workarounds and ways to rewire other software into your DAW.

Nonetheless, ghost notes can help with this timing issue. At the point you want natural sound, and nothing too rigid, thus unlocking your MIDI notes to move around freely should be done at this point. From this point on, naturally played ghost notes from natural playing or styles can be used to create well timed grooves, melodies, and big compositions. To naturally played piano, using these ghost notes, we can add bass to these "time spots" so as to sync well with the rhythm of the song. This is all just a starting point to learning about more advanced ways of creating music, but it is a good way to experiment with your own compositions. MIDI envelopes: These are used in many ways in studio recording, and often this is done with MIDI. We already saw ADSR, and this is essentially the basic idea behind audio envelopes. It is a way in which to control the outcome of audio in regard to volume fade in's and fade out's, as well as creative increases and decreases. Often times, in many pop type songs, growth in certain areas leads to a new musical part that creates relief or other changes. This can be created by using envelopes, but don't over use them. Many times people get caught up in this area and believe this is where you should focus your time. I do believe that it is an area to do well in, but in the same sense that you should learn to use dynamics well in your instrument playing. Good enveloping skills can lead to very interesting tracks, and it is just one of those things in the industry that is very powerful and quite responsible for the outcome of much success in music.

Psychology of Happiness

I cannot stress this enough. Over the years, I have struggled with personal happiness in relation to music and engineering. These have been my passions since I was a kid, and so it made me naturally happy. It took quite some time to realize the things that mattered in music, beyond all the music ideas and industry standards. Happiness is a science and is a big part of modern psychology. In fact, holistic type psychologies for happiness became popular from people like Rollo May and Carl Jung in recent times. In more recent times, the psychology of happiness started to make some sense, even to the degree of abstracting away for useful concepts. It is well known by psychologists these days that happiness comes from just being happy, and that means a handful of things. Expressing gratitude is good. This is because, in my opinion, when you realize what you are grateful for, you can see better the context of you in relation to the greater world, and it all happens with good, humble feelings about life and yourself. Just the act of smiling helps, too. This is just one of those things that parallels with the complexity of social life. Beyond these ideas, one of the biggest facts about all this that helps in our happiness, which scientists have studied at length, is simplifying life. Researchers now know that less complexity in life creates more happiness. Such as, going into a grocery store and instead of wondering around aimlessly among the thousands of products, finding the things you need and then getting on with your day. It has helped me, and I hope it helps you. I hope you go on to learn more, as this is very related to music in my opinion.

Builds, Risers, Stings

I wanted to talk about the two most significant aspects of professional music creation last, since they are both just that important: the best for last, right? Builds are the things that help make music sound professional. Whether they are enveloped risers in electronic music, drum fills on a rock track, or piano at the end of eight bars, builds help create a lot of interesting tension and excitement in music. I have always utilized the potential of builds to create interesting pieces. It is a creative process, and if you read through the book, especially about timing, you might know where I'm getting at here. When I was young, my builds were very elementary, but this was because I didn't know about timing and rhythm very well. I got better, but very slowly. In looking back at it all, the scenario was surely a situation with playing well, timing, and knowing styles and having good rhythm. It's a no brainer, and most students of music are aware of this. I was more of an audio nerd, thus my route was a bit different. If it isn't obvious enough, these components matter very much in the audio industry. This is why some artists are just better than others. Later in my young adult life, I learned how to create stings; that very final moment to songs and jingles that officially ends the track. I was very poor at it at first, but it got a little easier; especially when often all you need for most stings is your root note. From here, you can work with inversions, well thought-out ending masterpieces, or even percussion oriented endings. It is a vast landscape this part of music. Don't brush it off.

Music

This book is not about music theory, but I want to give a few tips to help in all this. I don't believe theory is everything, but I also don't believe no theory is that great either. In the world of music, you play or don't play. You are a producer, or a privileged musician, etc. I think playing is good, and I think playing simple songs is even better. It is too easy to get bogged down by complicated solos, piano or guitar licks, etc, but I believe mastering fundamentals and learning simple tunes is good, too. Chord progressions can go a long way in creating a good player. Scales that sprinkle over time become refined in the simplicity of music. It is a no brainer I believe, but I am surely no expert. I have been around music my whole life and have seen a lot of playing. I am fascinated by music, but I learned technology more than anything else. The best musicians are the one's that keep it simple and do things 'formally' in my opinion. I'll keep it at that. Here we are; the end. We made it through the

whole book! I hope it wasn’t too bad. I did my best to explain everything simply, but if you have any questions, feel free to email me at jgtovesonline@gmail.com. You can also check out my social media channels by just going into Google and typing in my name. Thanks for taking the time and I wish you well.
-Jeffrey Gates Toves

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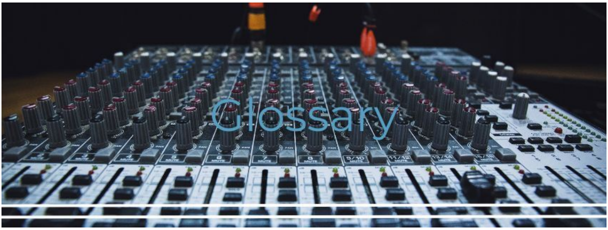

Glossary

Glossary

3D Audio - Sound that uses anatomically precise methods to direct sound to the human ear.

Amplifier - A process that adds power, in watts, to an audio signal to boost its volume.

Audio - The industry term for technologically processed sound.

Audio interface - A device for recording high quality audio into a computer.

Channel - A generic term used to indicate systems that allow for many similar processes to be worked with independently of eachother.

Condenser microphone - A microphone for recording very high quality sounds.

Compressor - A device that reduces peaks in an audio waveform.

Crossover - The technology used to divide an audio signal into a high part and a low part to be sent to tweeters and subwoofers.

DAT - Tape in digital format and was an early medium for sound reproduction.

Delay - An effect that repeats an original sound, sometimes in interesting ways.

Diaphragm - Parts to speakers and microphones that usually vibrate for the purpose of creating electrical equivalents.

Digital audio workstation - An a computer audio software that includes track recording and mixing.

Dynamic microphone - A microphone known for its durability and ability to withstand loud sounds.

Equalizer - A technology used to alter the frequencies of audio.

Flanger - A common audio effect.

Hard disk - An early form of tape recording.

Instrument - A device used to play music.

Keyboard - A device for playing samples of high quality instruments.

Limiter - A dynamic processing technology used for creating audio ceilings for various reasons.

MIDI - The main technology used in creating computer music and programming compositions.

Mini jack - A traditional jack in the audio industry; smaller than quarter inch.

Mixer - A technology for processing sound that aids in managing many inputs and inputs simultaneously.

Monitors - Studio speakers, or headphones.

Multitrack Recorder - A device for recording many overdubbs for professional purposes.

Pencil microphone - A condenser microphone housed in a small cabinet for recording in small spaces.

Phaser - A common audio effect.

Plugin - A term for software that is used in a modular way in DAW's to add external instruments or effects to a project.

Quarter inch jack - A common size jack in the audio industry, larger than a mini.

Recording - The process of recording sound into a recording technology.

Reel-to-reel - An early form of recording audio.

Reverb - An effects that simulates rooms and spaces.

Sequencer - A device used for looping sounds to create songs.

Shockmount - A holder for microphones which helps resist shaking and movement.

Subwoofer - The part of a speaker that processes low frequencies.

Surround sound - The process of delivering audio to more than two speakers.

Tape - An early medium of audio recording and playback that used magnetic particles.

Track - The term for a technology’s ability to record a track of audio.

Tweeter - The part of a speaker that processes the high frequencies.

Vinyl - The first form of audio recording, and still important in the modern audio industry.

XLR - A common cable in the audio industry that uses three pins to transmit high quality audio.

Chorus - A common audio effect.

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Info

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Notes

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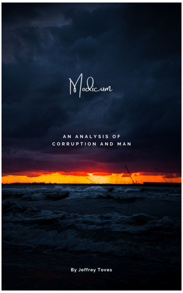
About the Author

Toves was born and raised in Ventura, CA, moving to Guam later in 1999. Toves first got his start as a teenager on Guam, experimenting with tapedecks and music machines, and graphics packages. He did radio jingles, remixing, recording, and promotions as a teenager, getting the support from local companies, which led him to engage in bigger media projects; production, graphic design, web design, and marketing. Toves started writing books in 2019 after leaving music for other pursuits. He is a member of the Authors Guild.

Also by Jeffrey Toves

Jeffrey Toves's books can be found on Amazon, Barnes & Noble, Kobo, and Smashwords.



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In this book, we stroll through the undercables of society through academia so as to attempt to get to grips with corruption in the world. We explore themes such as Maslow on corruption in history, philosophy and theology ideas, modern psychology, evolution, and tons more. Modicum is a short and sweet text in simply discussing corruption and the history of man in a modern context. There is no book like it, and it is a joy to read, as it also goes through the basics of evolution in biology and technology so as to become better aware of modern technological and scientific corruptions going on in the deep nets of complex society.